>BUSINESS MADE **SIMPLE**

NORTEL

Business Communication Manager R5.0 Configuration Guide:

MTS Allstream SIP Trunking Service

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1.0 Introduction

This configuration guide is to assist installers and administrators when connecting a BCM Release 5.0 system to MTS/Allstream's SIP Trunk service.

> Release 5.0 introduces a "common" software release for both BCM50 and BCM450. Thus, "BCM" will refer to BCM Release 5.0 and will not refer to any particular hardware platform (BCM50 or BCM450), unless otherwise stated.

This document does not describe procedures to configure the BCM for advanced functionality. For more information and procedures, please refer to the Nortel technical documentation found on the Nortel website at http://support.nortel.com.

1.1 Document Change History

Date Version	
December 17th 2009, Version 1.0	Original Publication

2.0 System Software

This section provides what system software versions were tested on the Nortel BCM50 Release 5.0 and MTS/Allstream's SIP Trunk Service.

Note: Nortel recommends installing the latest Smart Update (SU) Patch, which will contain previous patches and new fixes. The patches can be downloaded on the Nortel Technical Software Downloads Support site: http://support.nortel.com/go/main.jsp

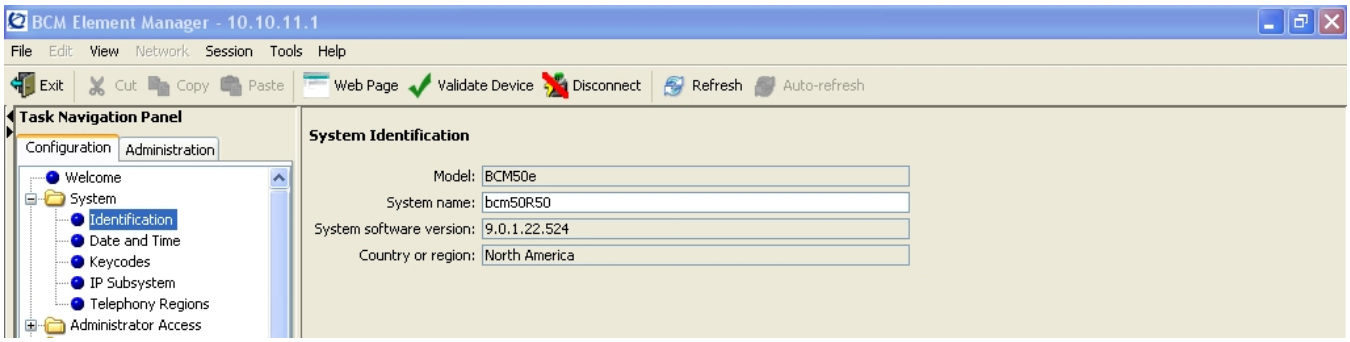
The BCM50 R5.0 was tested with BCM50 R5.0 SU Patch (BCM050 [1].R500.SU.System-001.200911-1) while connected to the MTS/Allstream SIP Trunk service.

To achieve successful interoperability between the BCM and MTS/Allstream SIP Trunk service, the various network elements must be running the version of software as shown below in Table 1:

System	Software/Loadware Version
Nortel BCM50	• **BCM R5.0** Base S/W: **9.0.1.00** • Patch: **BCM050.R500.SU.System-001.200911 or higher**
CS2000	• CS2K SN09U
Nextone SBC	• Rel v4.3m3

Table 1 Validated Equipment and Software

The Nortel Networks BCM must be running at least software version 9.0.1.22.XXX. You can check the version of BCM by viewing the following screen under ***Configuration → System → Identification***.



The following specifications have been considered:

SIP RFC3261

IP Codec: G711U and G729; Packet (frame) size: 20ms

DTMF: in band, RFC2833

Caller ID: P-Asserted-ID as per RFC3325 (Remote-Party-ID header supported but not recommended -deprecated-)

Caller ID blocking as per RFC3325 (Privacy id header)

Re-Invite to 0.0.0.0 or a=sendonly are supported for on-hold

G711 pass-through used for fax transmissions

Early-Media is supported

G729 is the preferred codec on Allstream side

3.0 Features

This section provides SIP Trunk features that are supported (or not supported) on the BCM Release 5.0 system with connected to MTS/Allstream's SIP Trunk services.

3.1 Network Features Supported

MTS Allstream provides the following network features:

- **G.729 CODEC Support in addition to G.711**
- **Survivability and Traffic Optimization Features**
    - Multi End Point Failover
    - Multi End Point Overflow
    - End Point Load Sharing
    - PRI Failover
- **Fax**
    - G.711 pass through is the only Fax protocol supported
- Multi CODEC Support (NRI&MRR)
    - G.729 (voice compression) in addition to G.711
    - G.729/ G.729a/ G.729b/ G.729a/b
    - Each G.729 DTE requires 45 kbps
        - G.711 DTE requires 100 kbps
    - Codec Negotiation
        - Requires customer's IP PBX to negotiate both Codecs
        - IP PBX must revert to G.711 when Fax/ Modem Tone is detected
        - IP PBX must be able to fallback on G.711 when receiving/ making calls to G.711 only customers (e.g. Cable co)

- **Basic 911 Service**
- **Alternate Number**
- **Alternate Trunk Group**
- **Inbound Overflow TN**
- **Inbound Overflow Trunk**
- **PSTN Connection**
- **Service Level Objectives**
- **Temporary Local Redirect Services**

3.2 Technical Caveats

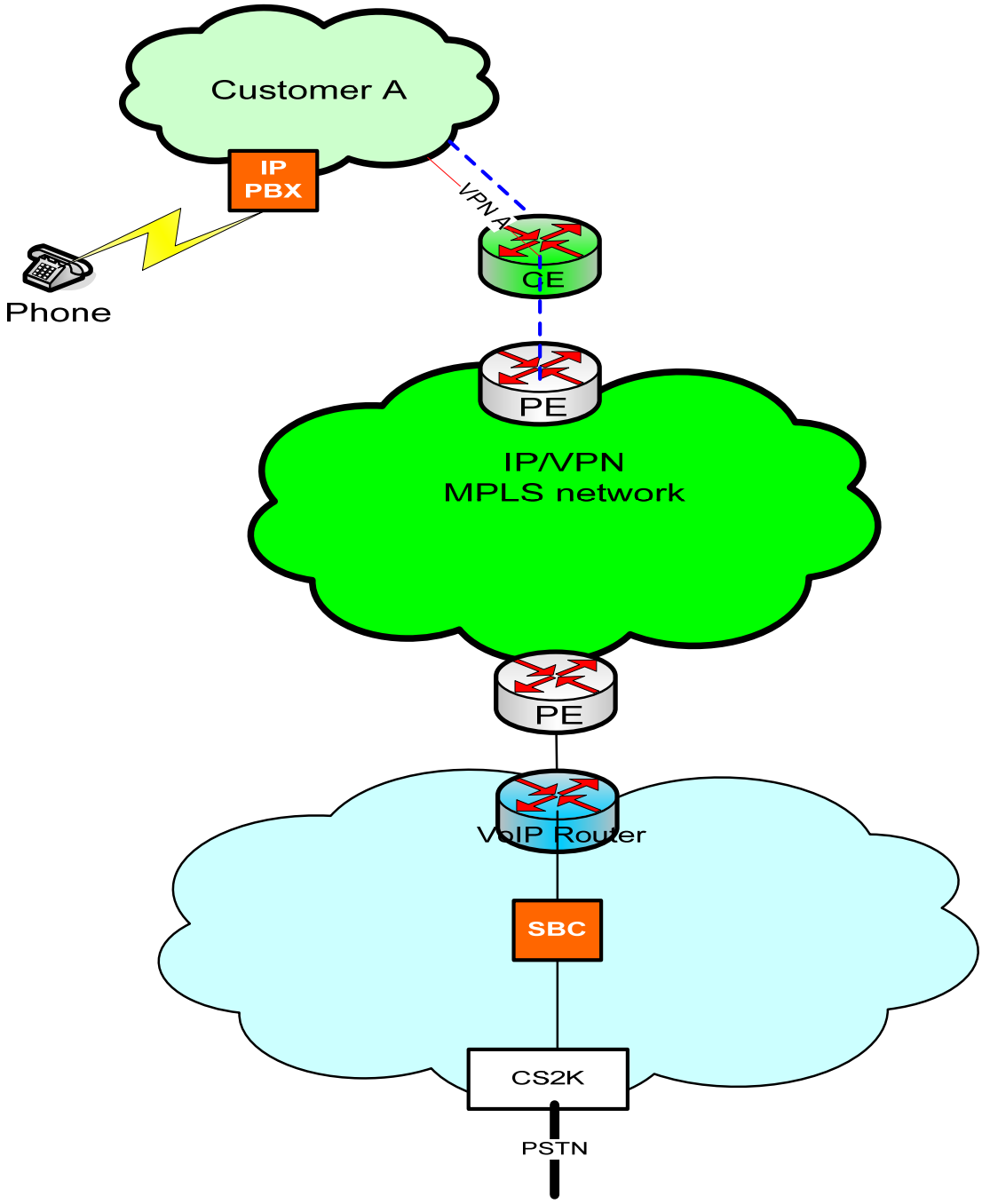
The following capabilities are **not** supported with this solution:

- T.38 fax is not supported
- Enhanced 911 is not supported
- 30ms packetization not supported

4.0 Network Diagram

The diagram below (Figure 1) shows the topology of the MTS/Allstream SIP Trunk service. In the diagram, there is no NAT between the BCM (IP PBX) and MTS/Allstream SIP Trunk service SBC.

Figure 1



5.0 System Configuration

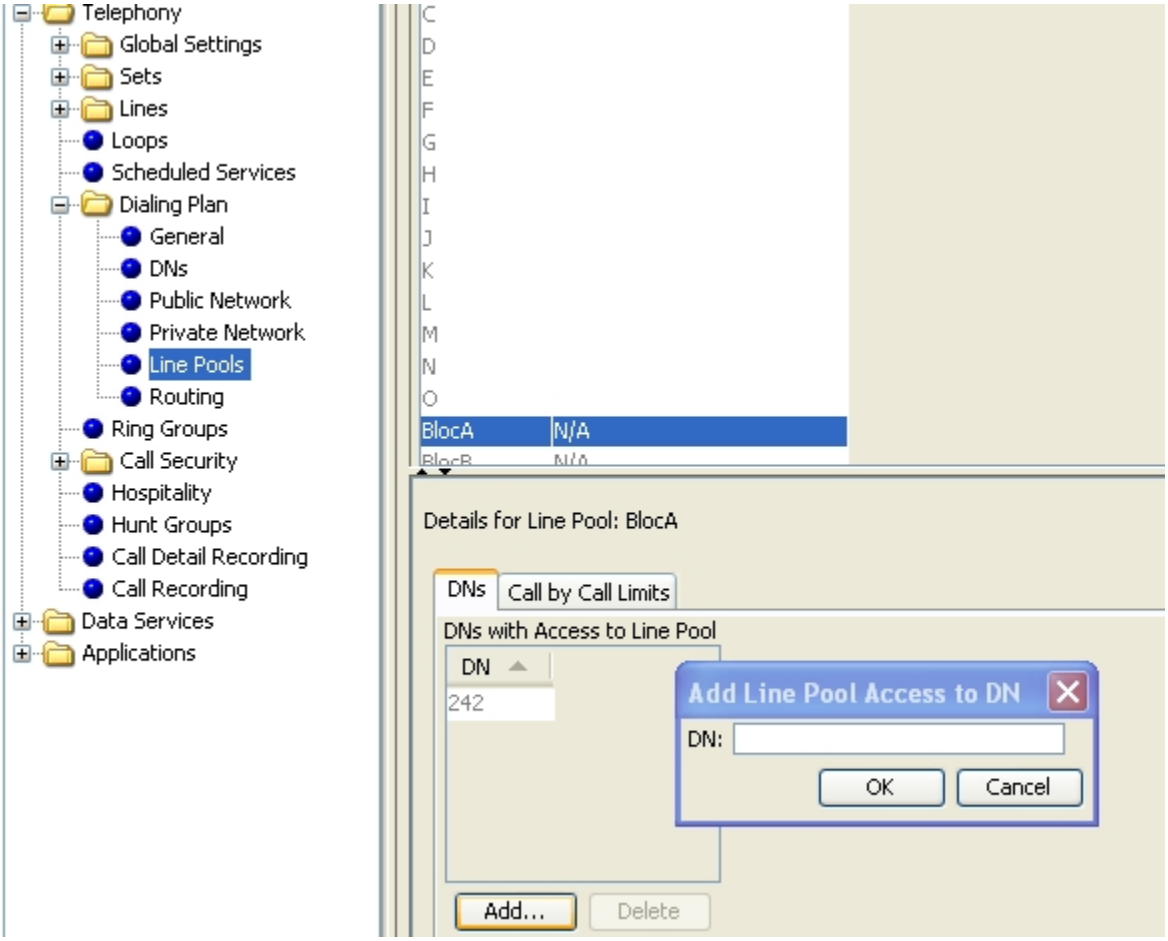
This section provides procedures for configuring a SIP Trunk on BCM Rls. 5.0 to MTS/Allstream SIP Trunk service.

5.1 BCM Configuration

In order to configure a SIP trunk between BCM Rls. 5.0 and MTS/Allstream do the following:

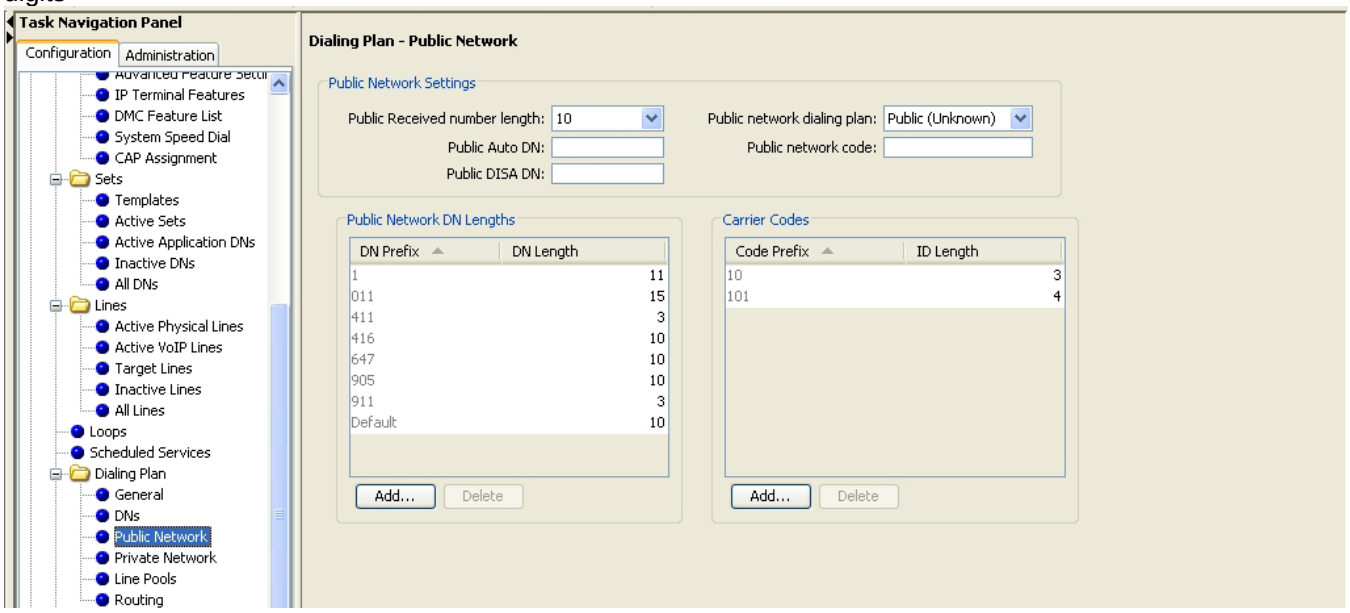
5.1.1 Line Pool Configuration

1. Under ***Configuration → Telephony → Dialing Plan***; select **Line Pools.**
2. Select **BlocA**.
3. Click on the "**Add**" button to add DNs of phones that need to access the above line pool.

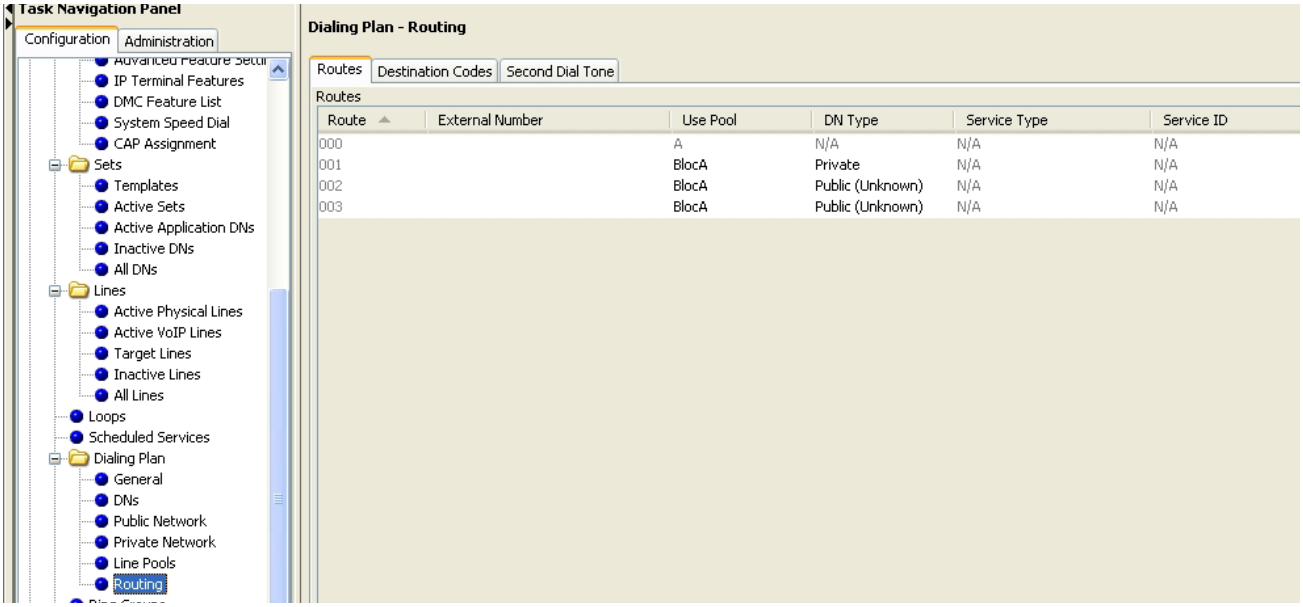


5.1.2 Dial Plan Configuration

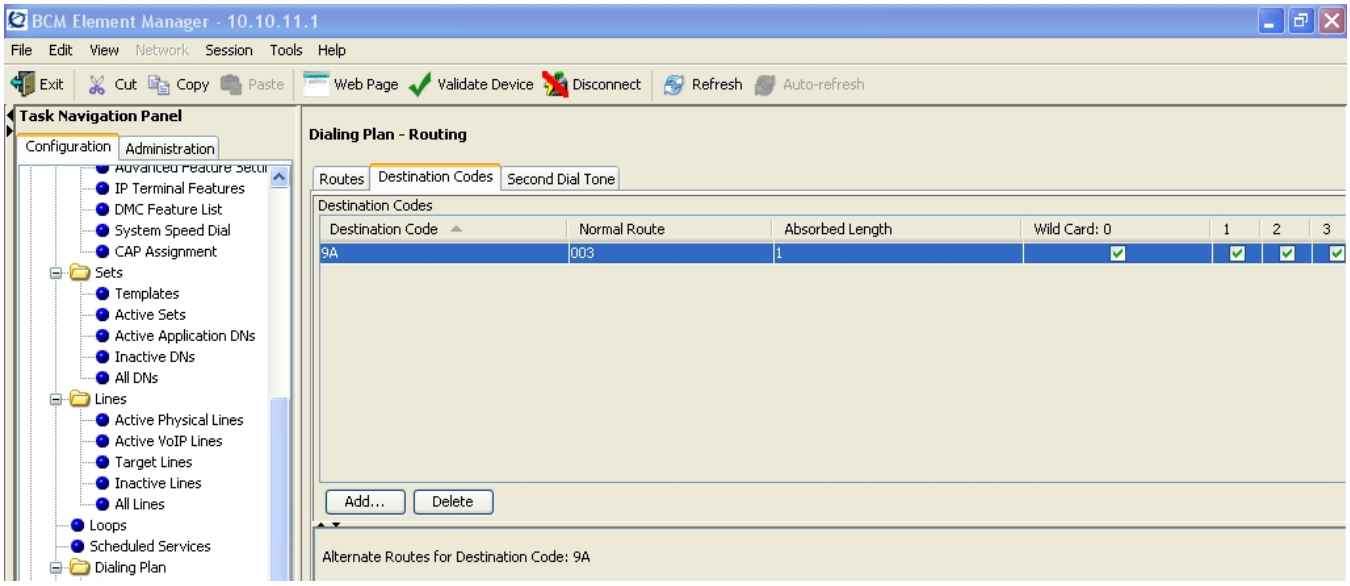
1. Under ***Configuration → Telephony → Dialing Plan → Public Network***, define the **Public Received** number length to "**10**"
digits



2. Set the **Public Network Dialing Plan** to **Public (Unknown)**
3. Under ***Configuration → Telephony → Dialing Plan → Routing*****,** and select the **Routes** tab
4. Add a route by clicking on the **Add** button.
5. In the **Add Route** dialog box, provide an unused route and click on the **OK** button.
6. The **Dialing Plan – Routing** table will be displayed.
7. Click on the route just created
8. Under the **Use Pool** column, double click to select **BlocA** from the drop down list.
9. Under the **DN Type** column, double click to select **Public (Unknown)** from the drop down list



10. Click on the **Destination Codes** tab.
11. Configure a destination code to route dialed digits by clicking on the **Add** button. Digits that begin with this destination code will be presented to the SIP trunking component on the BCM for routing towards the Service Provider.
12. In the **Add Destination Code** dialog box, enter a numeric number for the destination code and click on the **OK** button.
13. Select the row representing the Destination Code entered in the previous step
14. Under **Normal Route** column, double click and enter the route entered in Step 4.
15. Under the **Absorbed Length column**, specify the number of digits that will be absorbed before sending the rest of the digits to the service provider.



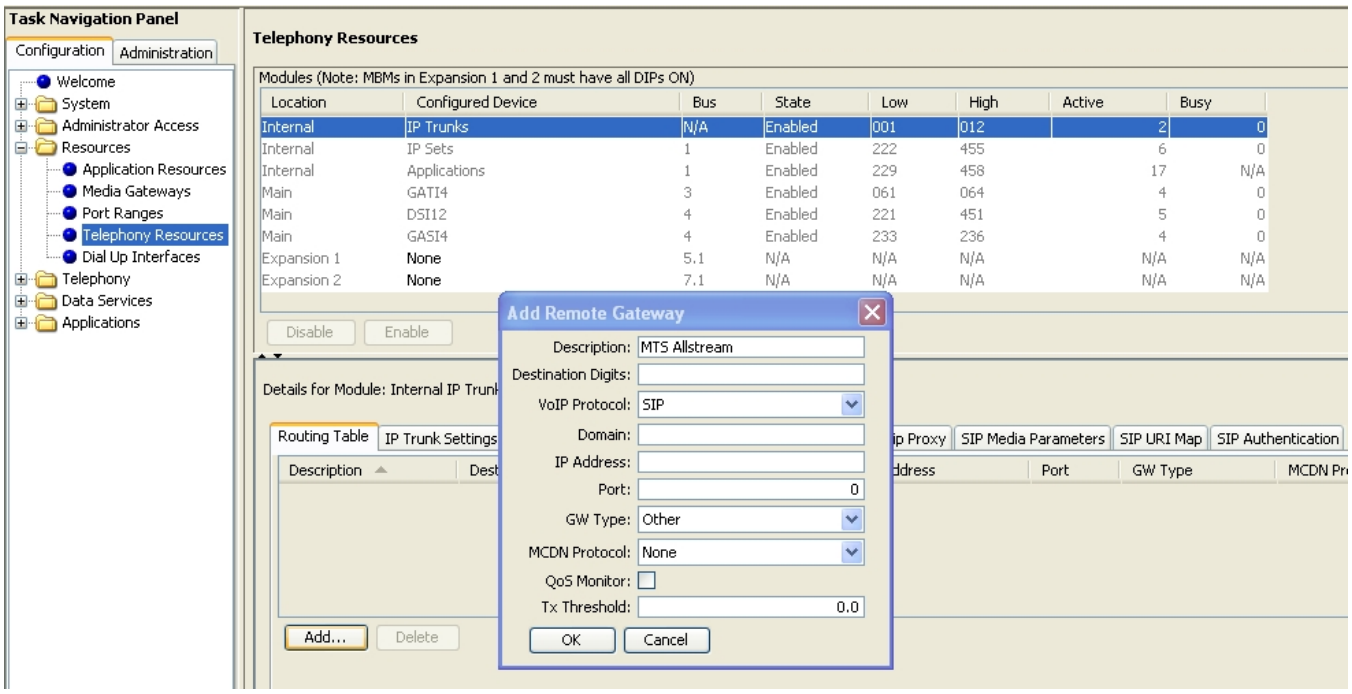
5.1.3 SIP Routing Table Configuration

There are two possible ways to configure the BCM to route outbound SIP calls to the Service Provider:

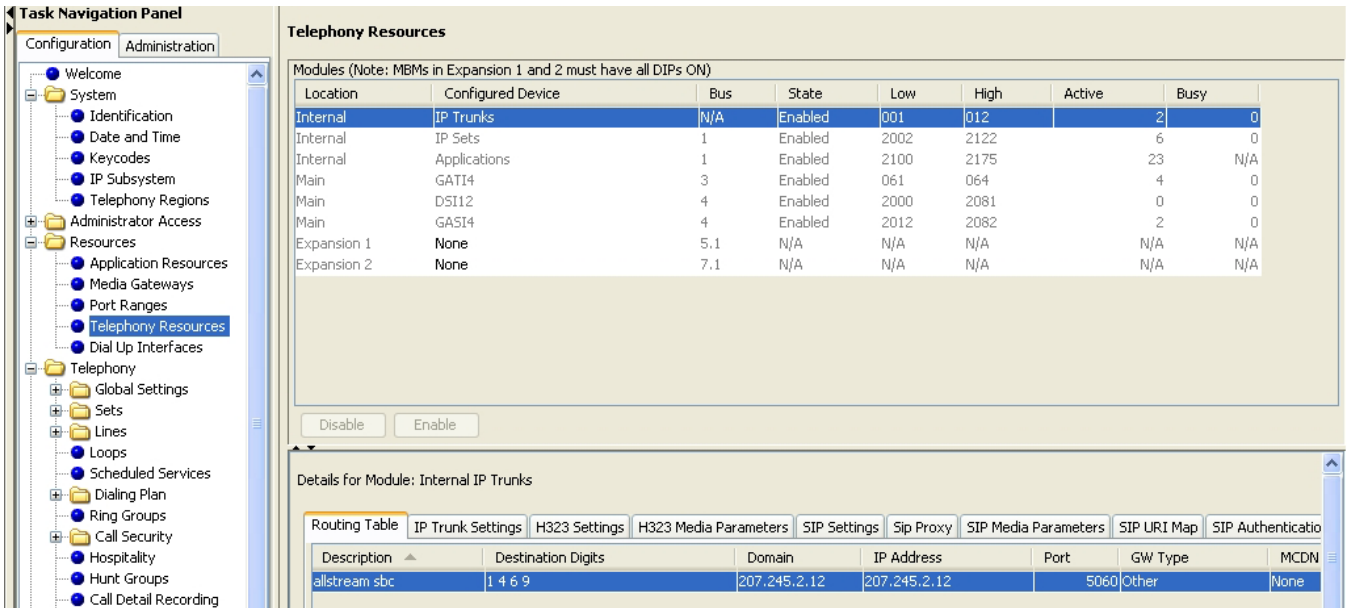
- Using the **Routing Table** tab found by clicking on IP Trunks module in ***Configuration → Resources → Telephony Resources.***
- Using the **SIP Proxy** tab found by clicking on IP Trunks module in ***Configuration → Resources → Telephony Resources.***

This guide illustrates how to do it using the Routing table. For details of how to use the SIP Proxy tab, see the BCM configuration guides.

1. Under ***Configuration → Resources → Telephony Resources;*** select module type **"IP Trunks"** and click on the "**Routing Table**" tab.

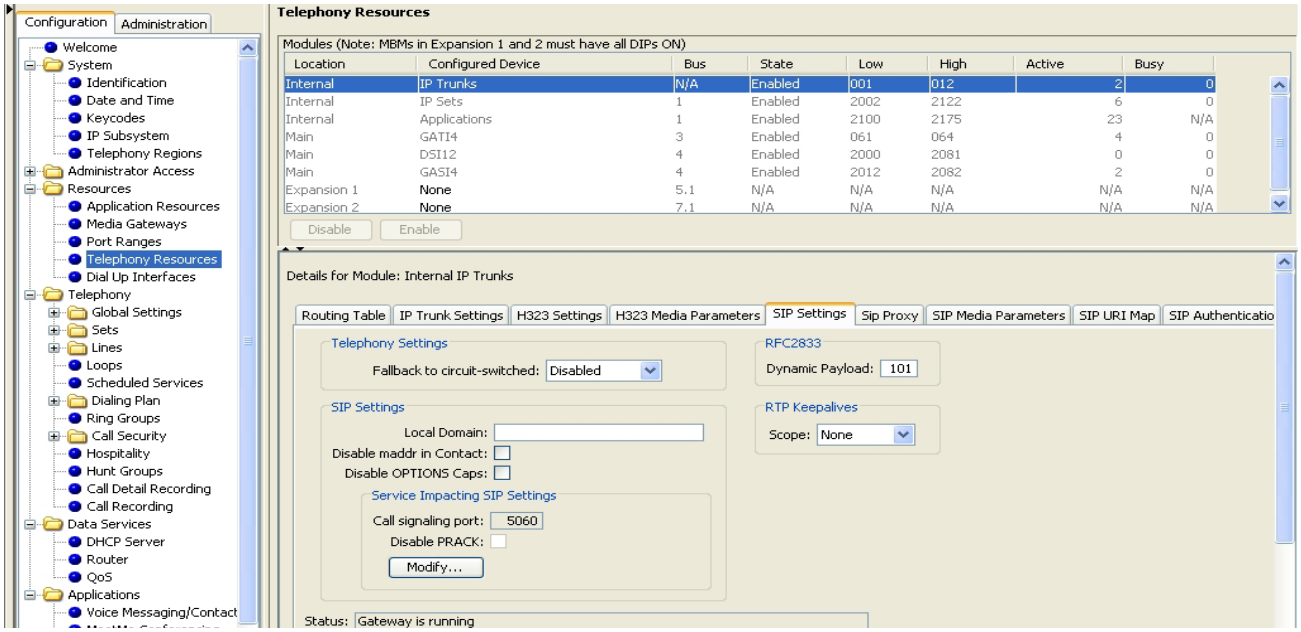


2. Add a "**Remote Gateway**" by clicking on the "**Add**" button.
3. In the "**Add Remote Gateway**" dialog box that is displayed, configure the fields as follows:
    - **Description**: Enter the logical name for the trunk destination
    - **Destination Digits**: Enter digits that identify the remote system as the call destination.
    - **VoIP Protocol**: SIP
    - **Domain**: SIP Domain name of MTS/Allstream. This will be provided by MTS/Allstream.
    - **IP Address**: Provide the IP address of the SBC. This will be provided by MTS/Allstream.
    - **Port**: Provide the optional UDP port number to which the BCM will send SIP messages.
    - **GW Type**: Set this field to **Other**.
    - **MCDN Protocol**: Set to **None**.
    - **QoS Monitor**: Leave unchecked
    - **Tx Threshold**: Leave this field at its default value of 0.0
4. Click the **OK** button


Task Navigation Panel
Configuration
Administration
Welcome
System
Identification
Date and Time
Keycodes
IP Subsystem
Telephony Regions
Administrator Access
Resources
Application Resources
Media Gateways
Port Ranges
Telephony Resources
Dial Up Interfaces
Telephony
Global Settings
Sets
Lines
Loops
Scheduled Services
Dialing Plan
Ring Groups
Call Security
Hospitality
Hunt Groups
Call Detail Recording
Telephony Resources
Modules (Note: MBMs in Expansion 1 and 2 must have all DIPs ON)
Location | Configured Device | Bus | State | Low | High | Active | Busy
Internal | IP Trunks | N/A | Enabled | 001 | 012 | 2 | 0
Internal | IP Sets | 1 | Enabled | 2002 | 2122 | 6 | 0
Internal | Applications | 1 | Enabled | 2100 | 2175 | 23 | N/A
Main | GATI4 | 3 | Enabled | 061 | 064 | 4 | 0
Main | DSI12 | 4 | Enabled | 2000 | 2081 | 0 | 0
Main | GASI4 | 4 | Enabled | 2012 | 2082 | 2 | 0
Expansion 1 | None | 5.1 | N/A | N/A | N/A | N/A | N/A
Expansion 2 | None | 7.1 | N/A | N/A | N/A | N/A | N/A
Disable
Enable
Details for Module: Internal IP Trunks
Routing Table
IP Trunk Settings
H323 Settings
H323 Media Parameters
SIP Settings
Sip Proxy
SIP Media Parameters
SIP URI Map
SIP Authenticatio
Description | Destination Digits | Domain | IP Address | Port | GW Type | MCDN
allstream sbc | 1 4 6 9 | 207.245.2.12 | 207.245.2.12 | 5060 | Other | None

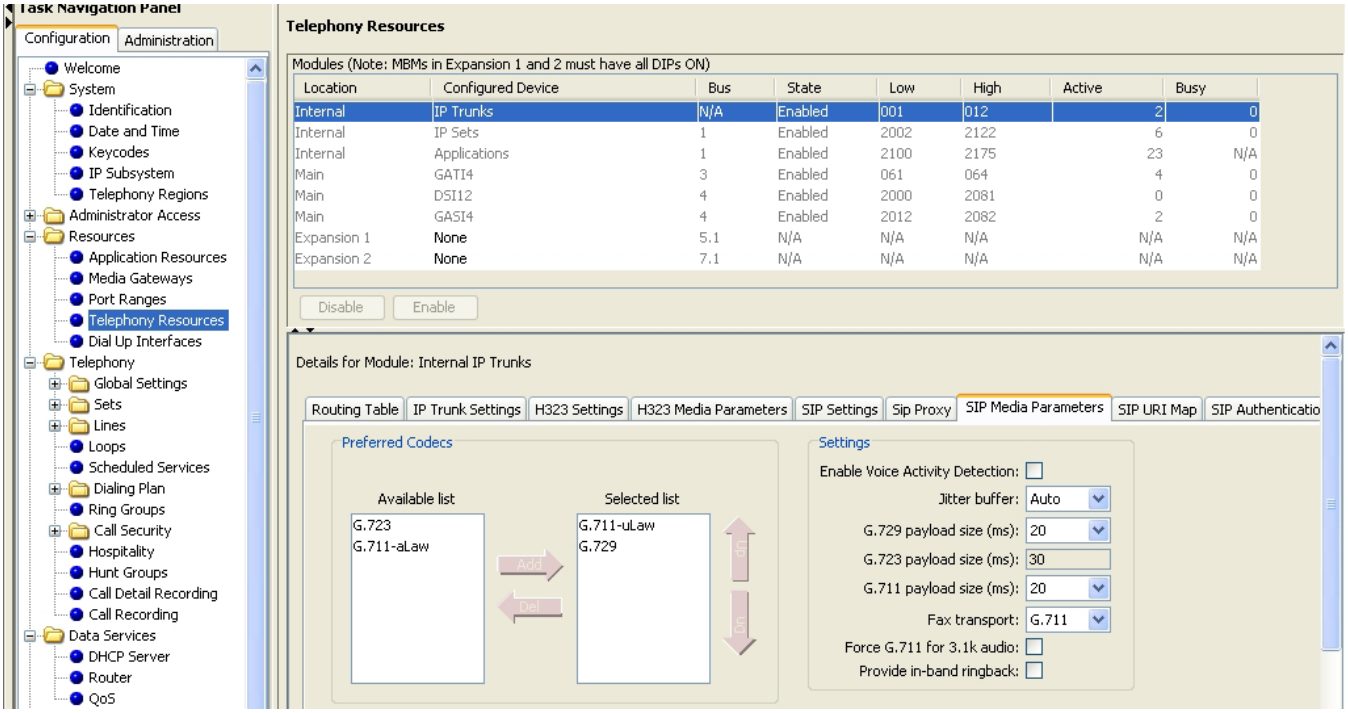
5.1.4 SIP Settings Configuration

1. Under ***Configuration → Resources → Telephony Resources:*** Select module type "**IP Trunks**" and click on the "**SIP Settings**" tab
2. In the **SIP Settings** sections:
   - **Local Domain**: Leave blank
   - **Disable maddr in Contact**: Leave **Un-Checked** to enable maddr in Contact header
   - **Disable OPTIONS Caps**: Leave **Un-Checked** to enable OPTIONS capability request by the BCM
   - **Dynamic Payload**: Enter **101** for MTS/Allstream SIP Trunk service
3. Leave the rest of parameters under the **SIP Settings** tab at the default values


Configuration
Administration
Telephony Resources
Modules (Note: MBMs in Expansion 1 and 2 must have all DIPs ON)
Location | Configured Device | Bus | State | Low | High | Active | Busy
Internal | IP Trunks | N/A | Enabled | 001 | 012 | 2 | 0
Internal | IP Sets | 1 | Enabled | 2002 | 2122 | 6 | 0
Internal | Applications | 1 | Enabled | 2100 | 2175 | 23 | N/A
Main | GATI4 | 3 | Enabled | 061 | 064 | 4 | 0
Main | DSI12 | 4 | Enabled | 2000 | 2081 | 0 | 0
Main | GASI4 | 4 | Enabled | 2012 | 2082 | 2 | 0
Expansion 1 | None | 5.1 | N/A | N/A | N/A | N/A | N/A
Expansion 2 | None | 7.1 | N/A | N/A | N/A | N/A | N/A
Disable
Enable
Details for Module: Internal IP Trunks
Routing Table
IP Trunk Settings
H323 Settings
H323 Media Parameters
SIP Settings
Sip Proxy
SIP Media Parameters
SIP URI Map
SIP Authenticatio
Telephony Settings
Fallback to circuit-switched: Disabled
RFC2833
Dynamic Payload: 101
SIP Settings
Local Domain:
Disable maddr in Contact:
Disable OPTIONS Caps:
RTP Keepalives
Scope: None
Service Impacting SIP Settings
Call signaling port: 5060
Disable PRACK:
Modify...
Status: Gateway is running

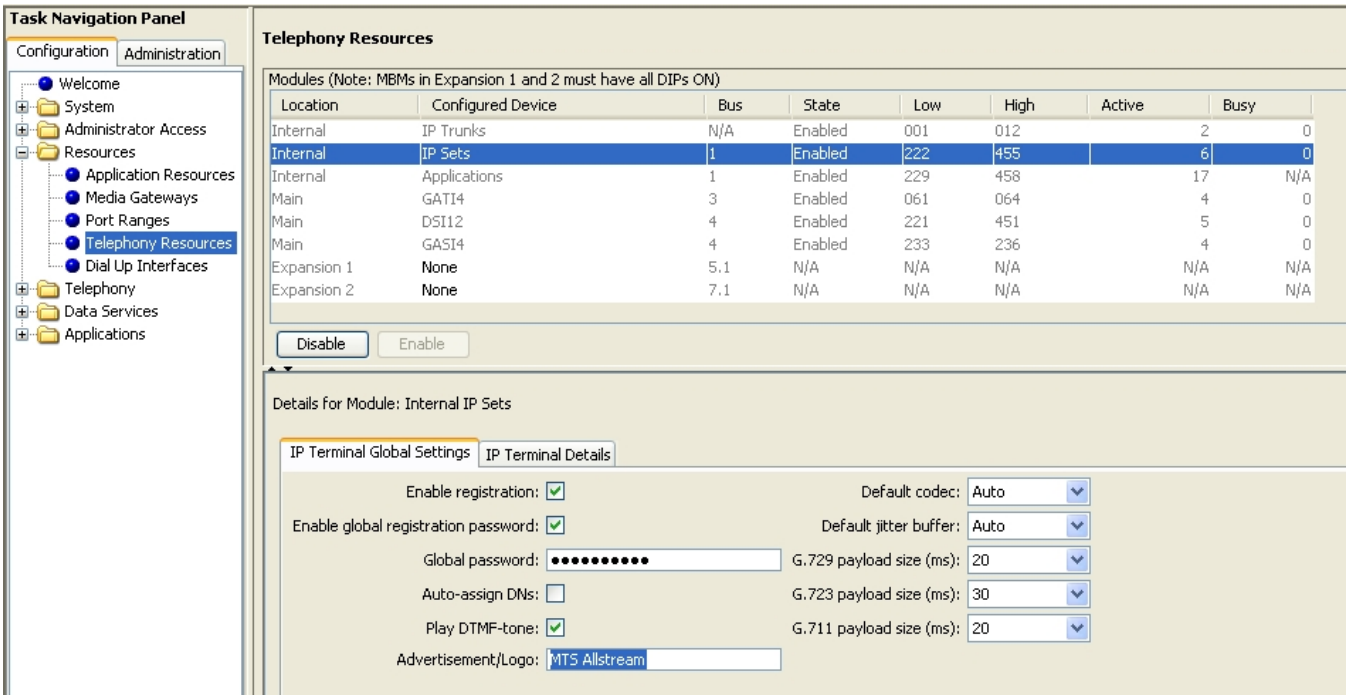
5.1.5 SIP Media Parameters Configuration

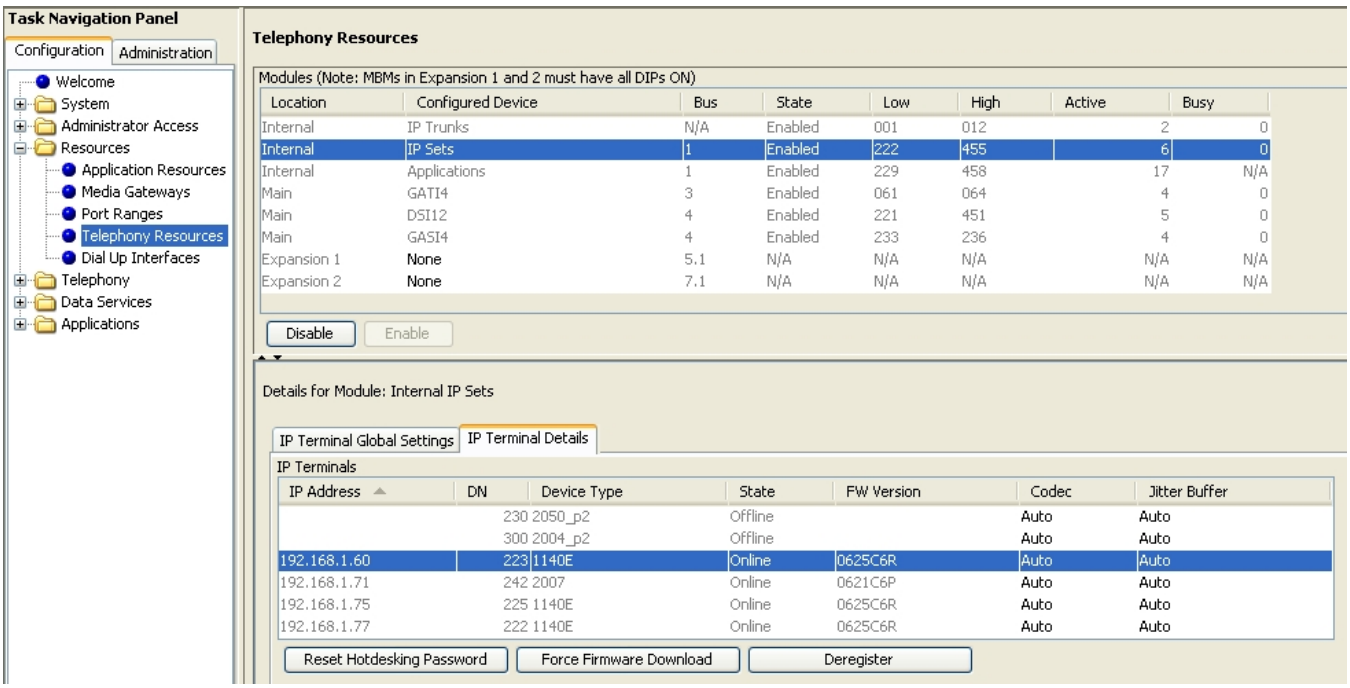
1. Under ***Configuration → Resources → Telephony Resources:*** Select module type "**IP Trunks**" and click on the "**SIP Media Parameters**" tab.
2. In the **Preferred Codecs** section, configure G.711-uLaw as the first preferred codec and G.729 as the second preferred codec.
3. In the codec **Settings** section, **Disable Enable Voice Activity Detection**.
4. In the codec **Settings** section, select **20ms** as the payload size for both G.729 and G.711
5. Again in the codec settings **Section** set **G.711** as the **Fax transport**.



5.1.6 IP Phones Media Parameters

1. Under ***Configuration → Resources → Telephony Resources***: Select module type "**IP Sets**" and click on "**IP Terminals Global Settings**".
2. Set the "**Default Codec**" for IP phones to **Auto** and set the payload size (ms) for G.729 and G.711 to **20ms** payload





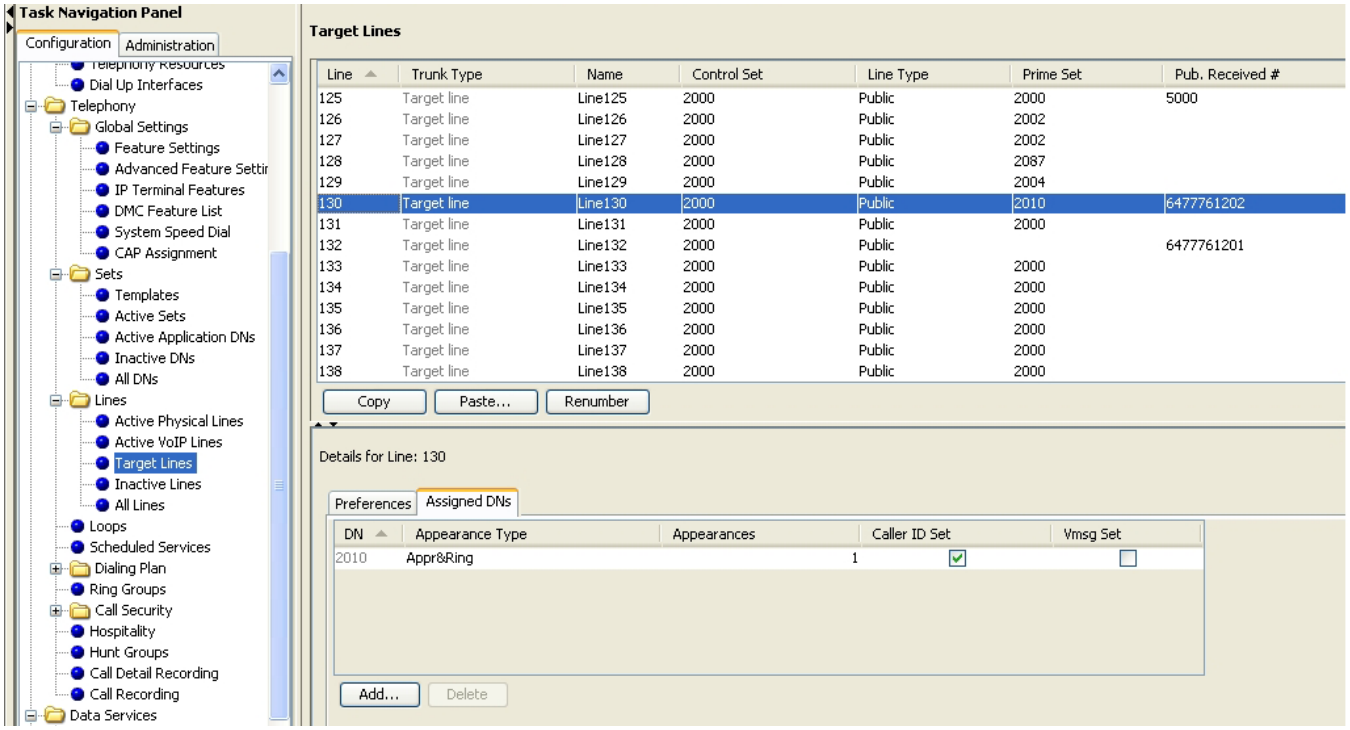
5.1.7 Configuring Incoming Calls from MTS/Allstream to BCM Rls. 5.0

This can be done in several ways;

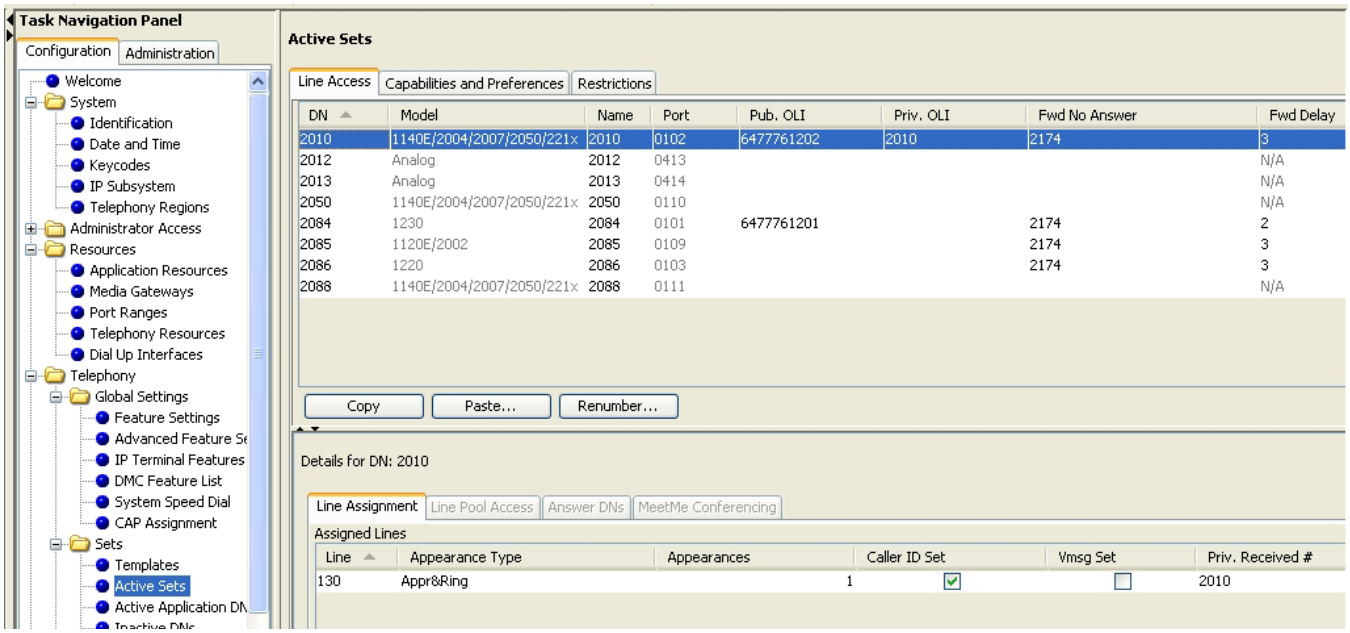
1. The DID associated with the MTS/Allstream SIP account assigned to the BCM can be associated with a “Target” line assigned to an individual phone or group of phone(s) and all calls to the DID will be routed to those phone(s) respectively.
2. All calls to the DID associated with the MTS/Allstream SIP account assigned to the BCM can be answered by the Auto Attendant (AA) and from there, a DN can be entered to reach phone on the BCM

5.1.7.1 Assigning DID to BCM Phones for Incoming Call

1. Navigate to ***Configuration → Telephony → Lines → Target Line*** and click on an unused target line
2. On the selected Target Line, set the "**Pub. Received #**" to the MTS/Allstream assigned DID.
3. Assign the DN of phones on the BCM that require an appearance on this target line. This will be the phones that will be alerted when call to the MTS/Allstream assign DID is received.
    a. Navigate to ***Configuration → Telephony → Lines → Target Line*** and click on the Target Line configured in Step 2 above
    b. Click on the **Assigned DNs** tab
    c. Click the **Add** button to add the DN of the phone(s) to this Target Line.



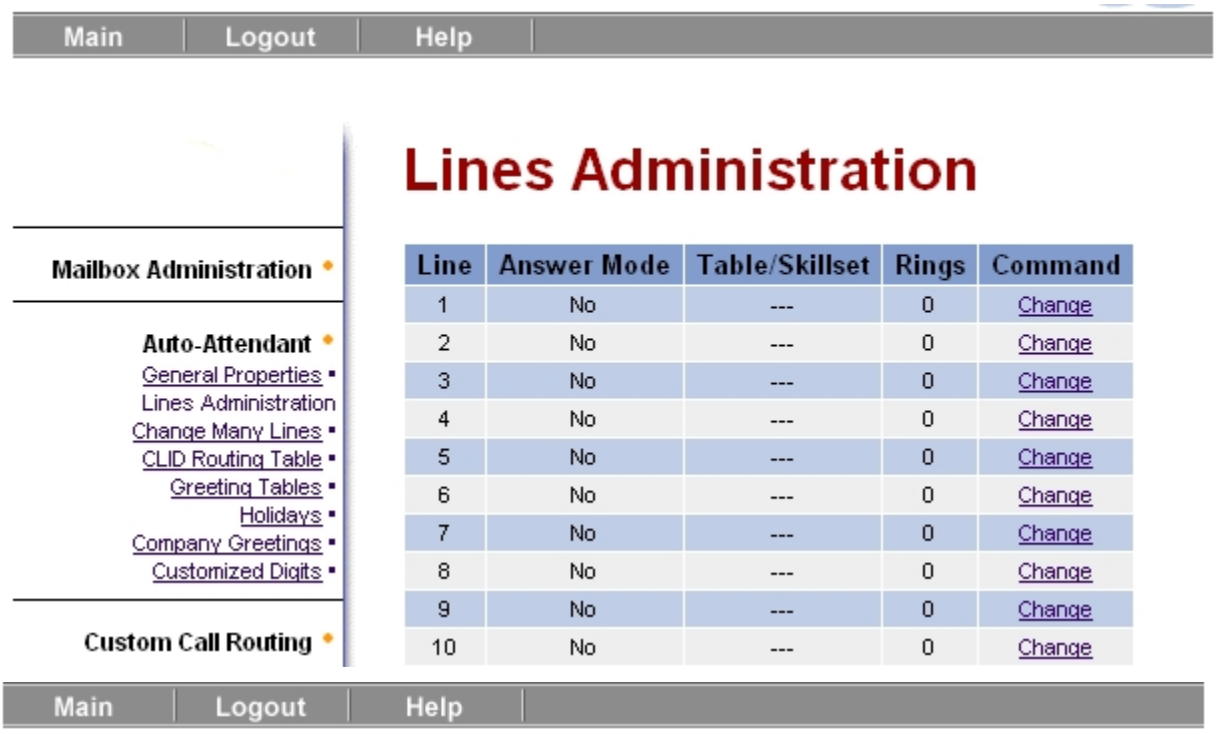
    d. OR Click on **Active Sets, Select DN, Line Access** tab, **Line Assignment** and Add Target Line to "**Appear & Ring**" and select Check box **Caller ID Set**.



5.1.7.2 Configuring AA to Answer Incoming Calls

Alternatively, the AA on the BCM can be configured to answer incoming calls and then call routed to a target phone on the BCM by entering the extension of the phone at the AA prompt. To do this,

1. Navigate to ***Configuration → Telephony → Lines → Target Line*** and click on an unused target line
2. On the selected Target Line, set the "**Pub. Received** #" to the 10-digits of the MTS/Allstream assigned DID.
3. Navigate to ***Configuration →Application →Voice Messaging/Contact Center***
4. Click on the **Launch CallPilot Manager**.
5. This launches a web browser to the BCM. Log in with the administrator credentials
6. On the left hand navigation menu, click **on Auto Attendant**
7. In the **Line Administration** web page, scroll down to the Target Line configured in Step 1.
8. Under the **Command Column**, click **Change**
9. In the Line Properties web page, select **Auto-Attendant** as the **Answer Mode**
10. Click the **Submit** button.


Main
Logout
Help
Lines Administration
Mailbox Administration
Auto-Attendant
General Properties
Lines Administration
Change Many Lines
CLID Routing Table
Greeting Tables
Holidays
Company Greetings
Customized Digits
Custom Call Routing
Line
Answer Mode
Table/Skillset
Rings
Command
1 No --- 0 Change
2 No --- 0 Change
3 No --- 0 Change
4 No --- 0 Change
5 No --- 0 Change
6 No --- 0 Change
7 No --- 0 Change
8 No --- 0 Change
9 No --- 0 Change
10 No --- 0 Change
Main
Logout
Help

Line Properties
Line Number: 5
Answer Mode: Auto-Attendant
Table/Skillset Number: 1
Number of Rings: 0
Submit
Cancel

6.0 References

The following are useful references to assist in this solution:

1. Partner Information Center:

Document Number: NN40020-201: SIP Interoperability Considerations and Technical Specifications for Business Communications Manager Release 5.0

2. BCM450 Technical Documentation",
   http://support.nortel.com/go/main.jsp?cscat=DOCUMENTATION&poid=19781
3. "BCM50 Technical Documentation",
   http://support.nortel.com/go/main.jsp?cscat=DOCUMENTATION&poid=15181

This document does not describe procedures to configure the BCM for advanced functionality. For more information and procedures, please refer to the Nortel technical documentation found on the Nortel website at http://support.nortel.com.

**********END**********

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